PCT

WORLD INTELLECTUAL PROPERTY ORGANIZATION
International Bureau

INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(51) International Patent Classification 6 :
H04R 25/00

A1

(11) International Publication Number: WO 95/08248

(43) International Publication Date: 23 March 1995 (23.03.95)

(21) International Application Number: PCT/US94/10419

(22) International Filing Date: 14 September 1994 (14.09.94)

(30) Priority Data:
08/123,503 17 September 1993 (17.09.93) US

(71) Applicant: AUDIOLOGIC, INCORPORATED [US/US]; Suite 200, 6655 Lookout Road, Boulder, CO 80301 (US).

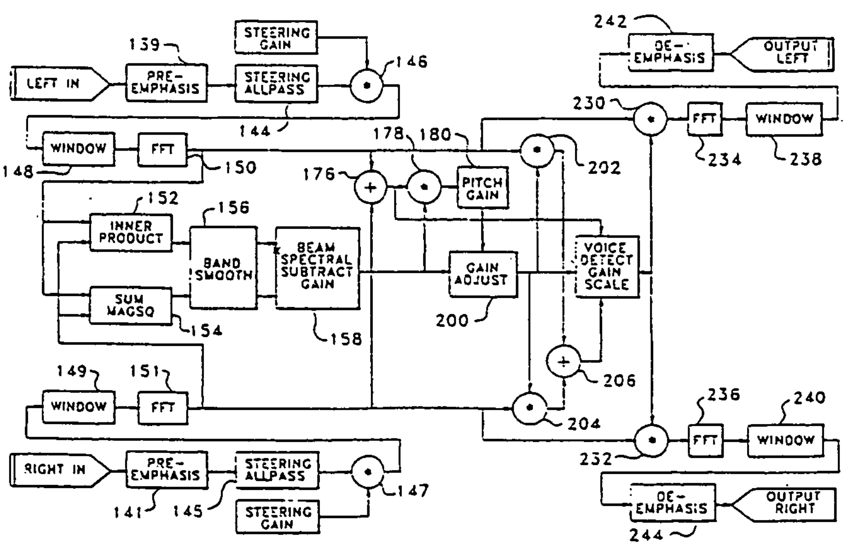
(72) Inventors: LINDEMANN, Eric; 2975 18th Street, Boulder, CO 80304 (US). MELANSON, John; 7973 Sagebrush Court, Boulder, CO 80301 (US).

(74) Agents: KNEARL, Homer, L. et al.; Hancock & Kneard, P.C., Suite 210, 3445 Penrose Place, Boulder, CO 80301 (US).

(81) Designated States: AM, AT, AU, BB, BG, BR, BY, CA, CH, CN, CZ, DE, DK, ES, FI, GB, GE, HU, JP, KE, KG, KP, KR, KZ, LK, LT, LU, LV, MD, MG, MN, MW, NL, NO, NZ, PL, PT, RO, RU, SD, SE, SI, SK, TJ, TT, UA, UZ, VN, European patent (AT, BE, CH, DE, DK, ES, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, ML, MR, NE, SN, TD, TG), ARIPO patent (KE, MW, SD).

Published
With international search report.

(54) Title: NOISE REDUCTION SYSTEM FOR BINAURAL HEARING AID



(57) Abstract

In this invention noise in a binaural hearing aid is reduced by analyzing the left and right digital audio signals to produce left and right signal frequency domain vectors and thereafter using digital signal encoding techniques to produce a noise reduction gain vector. The gain vector can then be multiplied against the left and right signal vectors to produce a noise reduced left and right signal vector. The cues used in the digital encoding techniques include directionality, short term amplitude deviation from long term average, and pitch. In addition, a multidimensional gain function based on directionality estimate and amplitude deviation estimate is used that is more effective in noise reduction than simply summing the noise reduction results of directionality alone and amplitude deviations alone. As further features of the invention, the noise reduction is scaled based on pitch-estimates and based on voice detection.

NOISE REDUCTION SYSTEM FOR BINAURAL HEARING AID

CROSS REFERENCE TO RELATED APPLICATIONS

The present invention relates to patent application entitled "Binaural Hearing Aid" Serial No.______________, filed September 17, 1993, which describes the system architecture of a hearing aid that uses the noise reduction system of the present invention.

BACKGROUND OF THE INVENTION

Field of the Invention:

This invention relates to binaural hearing aids, and more particularly, to a noise reduction system for use in a binaural hearing aid.

Description of Prior Art:

Noise reduction, as applied to hearing aids, means the attenuation of undesired signals and the amplification of desired signals. Desired signals are usually speech that the hearing aid user is trying to understand. Undesired signals can be any sounds in the environment which interfere with the principal speaker. These undesired sounds can be other speakers, restaurant clatter, music, traffic noise, etc. There have been three main areas of research in noise reduction as applied to hearing aids: directional beamforming, spectral subtraction, pitch-based speech enhancement.

The purpose of beamforming in a hearing aid is to create an illusion of "tunnel hearing" in which the listener hears what he is looking at but does not hear

of microphone sensors to be effective. This has made it difficult to incorporate these systems into practical hearing aid packages. One package that has been proposed consists of a microphone array across the top of eyeglasses {2}.

The frequency domain approaches which have been proposed {7, 8, 9} have performed better than delay and sum or adaptive filter approaches in reverberant listening environments and function with only two microphones. The problems related to the previously-published frequency domain approaches have been unacceptably long input to output time delay, distortion of the desired signal, spatial aliasing at high frequencies, and some difficulty in reverberant environments (although less than for the adaptive filter case).

While beamforming uses directionality to separate desired signal from undesired signal, spectral subtraction makes assumptions about the differences in statistics of the undesired signal and the desired signal, and uses these differences to separate and attenuate the undesired signal. The undesired signal is assumed to be lower in amplitude then the desired signal and/or has a less time varying spectrum. If the spectrum is static compared to the desired signal (speech), then a long-term estimation of the spectrum will approximate the spectrum of the undesired signal. This spectrum can be attenuated. If the desired speech spectrum is most often greater in amplitude and/or uncorrelated with the undesired spectrum, then it will pass through the system relatively undistorted despite attenuation of the undesired spectrum. Examples of work in spectral subtraction include references {11, 12, 13}.

referring to the complete written description of the preferred embodiments in conjunction with the following drawings.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 illustrates the preferred embodiment of the noise reduction system for a binaural hearing aid.

FIG. 2 shows the details of the inner product operation and the sum of magnitudes squared operation referred to in FIG. 1.

FIG. 3 shows the details of band smoothing operation 156 in FIG. 1.

FIG. 4 shows the details of the beam spectral subtract gain operation 158 in FIG. 1.

FIG. 5A is a graph of noise reduction gains as a serial function of directionality and spectral subtraction.

FIG. 5B is a graph of the noise reduction gain as a function of directionality estimate and spectral subtraction excursion estimate in accordance with the process in FIG. 4.

FIG. 6 shows the details of the pitch-estimate gain operation 180 in FIG. 1.

FIG. 7 shows the details of the voice detect gain scaling operation 208 in FIG. 1.

The two noise reduction systems can be connected back to back in serial fashion (e.g., beamformer followed by spectral subtractor). In this case, we can think in terms of a two-dimensional gain function of (D,STAD) with the function having a shape similar to that shown in FIG. 5A. With the serial connection, the gain function in FIG. 5A is rectangular. Values of (D,STAD) inside the rectangle generate a gain near unity which tends toward zero near the boundaries of the rectangle.

If we abandon the notion of a serial connection (beamformer followed by spectral subtractor) and instead think in terms of a general two-dimensional function of (D,STAD), then we can define non-rectangular gain contours, such as that shown in FIG. 5B Generalized Gain. Here we see that there is more interaction between the D and STAD values. A region which may have been included in the rectangular gain contour is now excluded because we are better able to take into consideration both D and STAD.

A common problem in spectral subtraction noise reduction systems is "musical noise". This is isolated bits of spectrum which manage to rise above the STAD threshold in discrete bursts. This can turn a steady state noise, such as a fan noise, into a fluttering random musical note generator. By using the combination of (D,STAD) we are able to make a better decision about a spectral component by insisting that not only must it rise above the STAD threshold, but it must also be reasonably on-line and that there is a continuous give and take between these two parameters.

Including f0 as a third cue gives rise to a three dimensional noise reduction system. We found it

transformed to the frequency domain, a directionality estimate can be made at each frequency point by comparing left and right values at each frequency. The directionality estimate is then used to generate a gain which is applied to the corresponding left and right frequency points and then the signals are resynthesized.

There are several key issues involved in the design of the basic analysis/synthesis system. In general, the analysis/synthesis system will treat the incoming signals as consecutive (possibly time overlapped) time segments of N sample points. Each N sample point segment will be transformed to produce a fixed length block of frequency domain coefficients. An optimum transform concentrates the most signal power in the smallest percentage of frequency domain coefficients. Optimum and near optimum transforms have been widely studied in signal coding applications {reference 19} where the desire is to transmit a signal using the fewest coefficients to achieve the lowest data rate. If most of the signal power is concentrated in a few coefficients, then only those coefficients need to be coded with high accuracy, and the others can be crudely coded or not at all.

The optimum transform is also extremely important for the beamformer. Assume that a signal consists of desired signal plus undesired noise signal. When the signal is transformed, some of the frequency domain coefficients will correspond largely to desired signal, some to undesired signal, and some to both. For the frequency coefficients with substantial contributions from both desired signal and noise, it is difficult to determine an appropriate gain. For frequency coefficients corresponding largely to desired signals the gain is near unity. For frequency coefficients

coefficients (or many bands of filtering) with energy concentrated in as few coefficients as possible (sharp transition bands between bandpass filters). Unfortunately, this kind of high frequency resolution implies large input sample segments which, in turn, implies long input to output delays in the system. In a hearing aid application, time delay through the system is an important parameter to optimize. If the time delay from input to output becomes too large (e.g. > about 40ms), the lips of speakers are no longer synchronized with sound. It also becomes difficult to speak since the sound of one's one voice is not synchronized with muscle movements. The impression is unnatural and fatiguing. A compromise must be made between input-output delay and frequency resolution. A good choice of analysis\synthesis architecture can ease the constraints on this compromise.

Another important consideration in the design of analysis/synthesis systems is edge effects. These are discontinuities that occur between adjacent output segments. These edge effects can be due to the circular convolution nature of fourier transform and inverse transforms, or they can be due to abrupt changes in frequency domain filtering (noise reduction gain, for example) from one segment to the next. Edge effects can sound like fluttering at the input segment rate. A well-designed analysis/synthesis system will eliminate these edge effects or reduce them to the point where they are inaudible.

The theoretical optimum transform for a signal of known statistics is the Karhoenen-Loeve Transform or KLT {19}. The KLT does not generally lend itself to practical implementation, but serves as a basis for

bandpass response with considerable leakage between bands so the coefficient energy concentration is poor. While an overlap-add scheme will guarantee smooth reconstruction in the case of convolution with a stationary finite impulse response of constrained length, when the impulse response is changing every block time, as is the case when we generate adaptive gains for a beamformer, then discontinuities will be generated in the output. It is as if we were to abruptly change all the coefficients in an FIR filter every block time. In an overlap-add system, the input to output minimum delay is:

$$D_{overlap\_add} = (1 + Z/2) * N + (\text{compute time for } 2*N \text{ FFT})$$

Where:

N = input segment length,

Z = number of zeros added to each block for zero padding.

A minimum value for Z is N, but this can easily be greater if the gain function is not sufficiently smooth over frequency. The frequency resolution of this system is N/2 frequency bins given conjugate symmetry of the transforms of the real input signal, and the fact that zero padding results in an interpolation of the frequency points with no new information added.

In the system design described in the preferred embodiments section of this patent, we use a windowed analysis/synthesis architecture. In a windowed FFT analysis/synthesis system, the input and output time domain sample segments are multiplied by a window function which in the preferred embodiment is a sine window for both the input and output segments. The frequency response of the bandpass filters (the transform

distortion. A hearing aid system for providing such low distortion left and right audio signals is described in the above-identified cross-referenced patent application entitled "Binaural Hearing Aid." The time domain digital input signal from each ear is passed to one-zero pre-emphasis filters 139, 141. Pre-emphasis of the left and right ear signals using a simple one-zero high-pass differentiator pre-whitens the signals before they are transformed to the frequency domain. This results in reduced variance between frequency coefficients so that there are fewer problems with numerical error in the fourier transformation process. The effects of the preemphasis filters 139, 141 are removed after inverse fourier transformation by using one-pole integrator deemphasis filters 242 and 244 on the left and right signals at the end of noise reduction processing. Of course, if binaural compression follows the noise reduction stage of processing, the inverse transformation and deemphasis would be at the end of binaural compression.

This preemphasis/deemphasis process is in addition to the preemphasis/deemphasis used before and after radio frequency transmission. However, the effect of these separate preemphasis/deemphasis filters can be combined. In other words, the RF received signal can be left preemphasized so that the DSP does not need to perform an additional preemphasis operation. Likewise, the output of the DSP can be left preemphasized so that no special preemphasis is needed before radio transmission back to the ear pieces. The final deemphasis is done in analog at the ear pieces.

In FIG. 1, after preemphasis, if used, the left and right time domain audio signals are passed through

samples 128..383, the third spans samples 256..511, etc. The processing of each consecutive block is identical.

The noise reduction processing begins by multiplying the left and right 256 point sample blocks by a sine window in operations 148, 149. A fast Fourier transform (FFT) operation 150, 151 is then performed on the left and right blocks. Since the signals are real, this yields a 128 point complex frequency vector for both the left and right audio channels. The elements of the complex frequency vectors will be referred to as bin values. So there are 128 frequency bins from F=0 (DC) to F=Fsamp/2 Khz.

The inner product of, and the sum of magnitude squares of each frequency bin for the left and right channel complex frequency vector, is calculated by operations 152 and 154, respectively. The expression for the inner product is:

Inner Product(k) = Real(Left(k))*Real(Right(k)) + Imag(Left(k))*Imag(Right(k)

and is implemented, as shown in FIG. 2. The operation flow in FIG. 2 is repeated for each frequency bin. On the same FIG. 2, the sum of magnitude squares is calculated as:

Magnitude Squared Sum(k) = Real(Left(k))^2 + Real(Right(k))^2 + Imag(Left(k))^2 + Imag(Right(k)^2.

An inner product and magnitude squared sum are calculated for each frequency bin forming two frequency domain vectors. The inner product and magnitude squared

Spatial aliasing occurs when the wave lengths of signals arriving at the left and right ears are shorter than the space between the ears. When this occurs, a signal arriving from off-axis can appear to be perfectly in-phase with respect to the two ears even though there may have been a K*2*PI (K some integer) phase shift between the ears. Axis in "off-axis" refers to the centerline perpendicular to a line between the ears of the user; i.e., the forward direction from the eyes of the user. This spatial aliasing phenomenon occurs for frequencies above approximately 1500 Hz. If the real world, signals consist of many spectral lines, and at high frequencies these spectral lines achieve a certain density over frequency -- this is especially true for consonant speech sounds -- and if the estimate of directionality for these frequency points are averaged, an on-axis signal continues to appear on-axis. However, an off-axis signal will now consistently appear off-axis since for a large number of spectral lines, densely spaced, it is impossible for all or even a significant percentage of them to have exactly integer K*2*PI phase shifts.

The inner product average and magnitude squared sum average vectors are then passed from the band smoother 156 to the beam spectral subtract gain operation 158. This gain operation uses the two vectors to calculate a gain per frequency bin. This gain will be low for frequency bins, where the sound is off-axis and/or below a spectral subtraction threshold, and high for frequency bins where the sound is on-axis and above the spectral subtraction threshold. The beam spectral subtract gain operation is repeated for every frequency bin.

Alternatively, the averages could be eliminated and instead the resulting estimate d could be averaged, but this is not the preferred embodiment. In fact, this alternative is not a good choice. By averaging inner product and magnitude squared sum independently, small magnitudes contribute little to the "d" estimate. Without preliminary smoothing, large changes in d can result from small magnitude frequency components and these large changes contribute unduly to the d average.

The magnitude square sum average is passed through a long-term averaging filter 170, which is a one pole filter with a very long time constant. The output from one pole smoothing filter 162, which smooths the magnitude square sum is subtracted at operation 172 from the long term average provided by filter 170. This yields an excursion estimate value representing the excursions of the short-term magnitude sum above and below the long term average and provides a basis for spectral subtraction. Both the direction estimate and the excursion estimate are input to a two dimensional lookup table 174 which yields the beam spectral subtract gain.

The two-dimensional lookup table 174 provides an output gain that takes the form shown in FIG. 5B. The region inside the arched shape represents values of direction estimate and excursion for which gain is near one. At the boundaries of this region, the gain falls off gradually to zero. Since the two-dimensional table is a general function of directionality estimate and spectral subtraction excursion estimate, and since it is implemented in read/write random access memory, it can be modified dynamically for the purpose of changing beamwidths.

harmonic grid related to F_0 is selected from table 186 by operation 192 and used to form the pitch gain. Multiply operation 194 produces the F_0 harmonic grid scaled by the pitch confidence measure. This is the pitch gain vector.

In FIG. 1, both pitch gain and beam spectral subtract gain are input to gain adjust operation 200. The output of the gain adjust operation is the final per frequency bin noise reduction gain. For each frequency bin, the maximum of pitch gain and beam spectral subtract gain is selected in operation 200 as the noise reduction gain.

Since the pitch estimate is formed from the partially noise reduced signal, it has a strong probability of reflecting the pitch of the desired signal. A pitch estimate based on the original noisy signal would be extremely unreliable due to the complex mix of desired signal and undesired signals.

The original frequency domain left and right ear signals from FFTs 150 and 151 are multiplied by the noise reduction gain at multiply operations 202 and 204. A sum of the noise reduced signals is provided by summing operation 206. The sum of noise reduced signals from summer 206, the sum of the original non-noise reduced left and right ear frequency domain signals from summer 176, and the noise reduction gain are input to the voice detect gain scale operation 208 shown in detail in FIG. 7.

In FIG. 7, the voice detect gain scale operation begins by calculating, at operation 210, the ratio of the total power in the summed left and right noised reduced signals to the total power of the summed left and right

signals. The left and right ear noise reduced frequency domain signals are then inverse transformed at FFTs 234 and 236. The resulting time domain segments are windowed with a sine window and 2:1 overlap-added to generate a left and right signal from window operations 238 and 240. The left and right signals are then passed through deemphasis filters 242, 244 to produce the stereo output signal. This completes the noise reduction processing stage.

While a number of preferred embodiments of the invention have been shown and described, it will be appreciated by one skilled in the art, that a number of further variations or modifications may be made without departing from the spirit and scope of my invention.

References Cited In Specification:

1. Evaluation of an adaptive beamforming method for hearing aids. J. Acoustic Society of America 91(3). Greenberg, Zurek.

2. Improvement of Speech Intelligibility in Noise: Development and Evaluation of a New Directional Hearing Instrument Based on Array Technology. Thesis from Delft University of Technology. Willem Soede

3. Multimicrophone adaptive beamforming for interference reduction in hearing aids. Journal of Rehabilitation Research and Development, Vol. 24 No. 4. Peterson, Durlach, Rabinowitz, Zurek.

4. Multimicrophone signal processing technique to remove room reverberation from speech signals. J. Acoustic Society of America 61. Allen, Berkley, Blauert.

14. Gaik (1990): Untersuchungen zur binaurelen Verarbeitung kopfbesogener Signale. Fortschr.-Be. VDI Reihe 17 Nr. 63. Dusseldorf: VDI-Verlag.

15. Lindemann W. (1986): Extension of a binaural cross-correlation model by contralateral inhibition. I. Simulation of lateralization of stationary signals. JASA 80, 1608-1622.

16. Openheim and Schaefer. (1989) Discrete-Time Signal Processing. Prentice Hall.

17. Parsons (1976) Separation of speech from interfering speech by means of harmonic selection. JASA 60 911-918

18. Stubbs, Summerfield (1988) Evaluation of two voice-separation algorithms using normal-hearing and hearing-impaired listeners. JASA 84 (4) Oct. 1988

19 Jayant, Noll. (1984) Digital coding of waveforms. Prentice-Hall.

20. Crochiere, Rabiner. (1983) Multirate Digital Signal Processing. Prentice-Hall

21 Malvar (1992) Signal Processing With Lapped Transforms, Artech House, Norwood MAS, 1992

22. Vaidyanathan (1993) Multirate Systems and Filter Banks, Prentice-Hall

23. Daubauchies (1992) Ten Lectures On Wavelets, SIAM CBMS seties, April 1992

What is claimed is:

3. In a binaural hearing aid system having left and right digital audio time domain signals, apparatus for reducing noise in the left and right audio signals comprising:

means for analyzing the left and right audio signals into frequency domain vectors;

means for applying signal encoding techniques based on cues derived from the left and right audio vectors to provide a noise reduction gain vector;

means for adjucting the left and right audio signal vectors with the noise reduction gain vector to reduce the noise in the left and right audio vectors; and

means for synthesizing left and right time domain digital audio signals from the noise reduce left and right audio vectors.

4. The system of claim 3 wherein the cues in said applying means include directionality, short term amplitude deviation from long term average, and pitch.

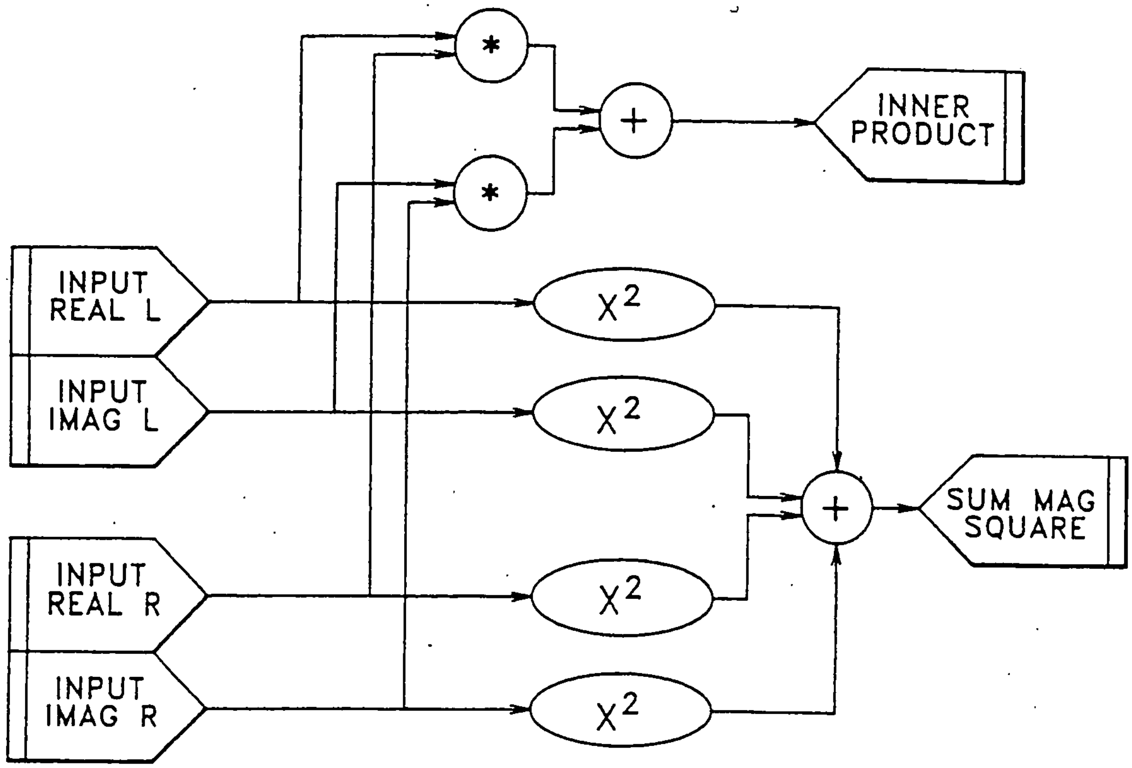


FIG.2

NOTE: THIS CIRCUIT IS REPEATED FOR EVERY FREQUENCY F OF THE FFT

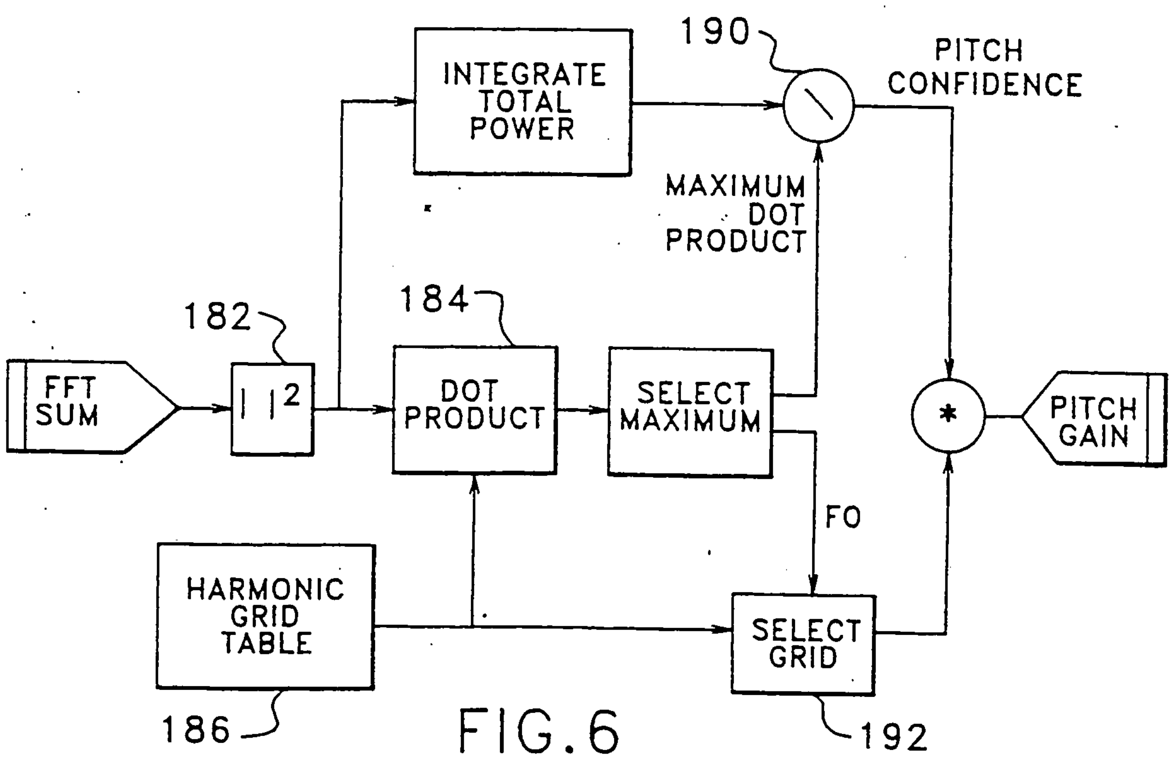


FIG.6

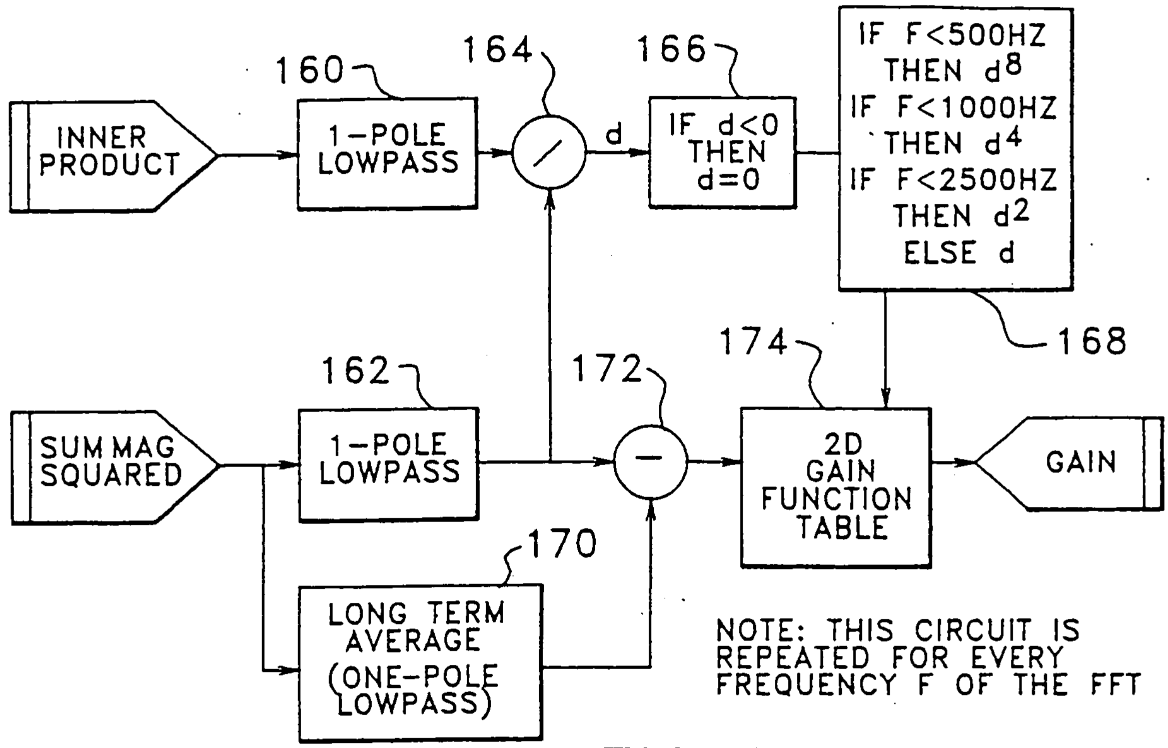

160
164
166
INNER PRODUCT
1-POLE LOWPASS
d
IF d<0 THEN d=0
IF F<500HZ THEN d8
IF F<1000HZ THEN d4
IF F<2500HZ THEN d2
ELSE d
168
174
162
172
SUM MAG SQUARED
1-POLE LOWPASS
2D GAIN FUNCTION TABLE
GAIN
170
LONG TERM AVERAGE (ONE-POLE LOWPASS)
NOTE: THIS CIRCUIT IS REPEATED FOR EVERY FREQUENCY F OF THE FFT

FIG.4

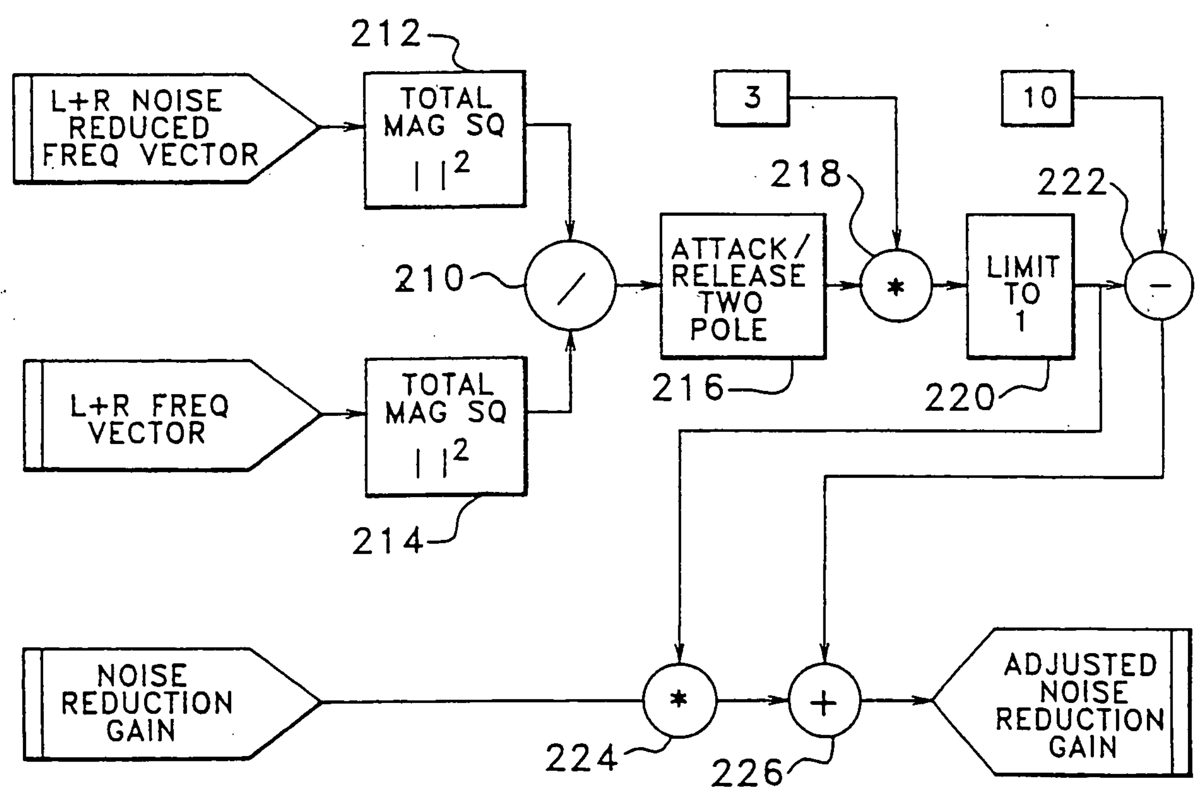

212
L+R NOISE REDUCED FREQ VECTOR
TOTAL MAG SQ | |2
3
10
218
222
210
ATTACK/ RELEASE TWO POLE
LIMIT TO 1
L+R FREQ VECTOR
TOTAL MAG SQ | |2
216
220
214
NOISE REDUCTION GAIN
ADJUSTED NOISE REDUCTION GAIN
224
226

FIG.7

INTERNATIONAL SEARCH REPORT

International Application No
PCT/US 94/10419

A. CLASSIFICATION OF SUBJECT MATTER
IPC 6 H04R25/00

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 6 H04R G10L H04S H03G G01R H04B G01S

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	WO,A,92 08330 (COCHLEAR PTY.) 14 May 1992 see page 1, line 1-5 see page 7, line 24 - page 10, line 12 see page 14, line 24 - page 3, line 9 ---	1-4
A	J.ACOUST.SOC.OF AMERICA, vol.93, no.4, April 1993, WASHINGTON DC pages 1851 - 1865, XP000363366 COLLINS ET AL. 'APPLICATIONS OF OPTIMAL TIME-DOMAIN BEAMFORMING.' see paragraph 1-8 ---	1,2
A	GB,A,2 238 696 (C.S.C.) 5 June 1991 see page 6, line 23 - page 8, line 25 see page 15, line 13 - page 22, line 33 see page 25, line 1 - page 35, line 8 --- -/--	1,3

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

* Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

2

Date of the actual completion of the international search	Date of mailing of the international search report
30 November 1994	05.01.95

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax (+31-70) 340-3016	Zanti, P

Form PCT/ISA/210 (second sheet) (July 1992)

INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No
PCT/US 94/10419

Patent document cited in search report	Publication date	Patent family member(s)		Publication date
WO-A-9208330	14-05-92	CA-A-	2095344	02-05-92
		EP-A-	0555278	18-08-93
		JP-T-	6506322	14-07-94
GB-A-2238696	05-06-91	US-A-	5307441	26-04-94
		AU-B-	6485894	01-09-94
		AU-B-	652134	18-08-94
		AU-A-	6707490	06-06-91
		CA-A,C	2031006	30-05-91
		JP-A-	3211599	17-09-91
US-A-4646254	24-02-87	NONE		
WO-A-8906877	27-07-89	CA-A-	1332626	18-10-94
		DE-D-	68913139	24-03-94
		DE-T-	68913139	23-06-94
		EP-A-	0367803	16-05-90
		GB-A-	2220330	04-01-90
		JP-T-	2503256	04-10-90
		US-A-	5133013	21-07-92
EP-A-0276159	27-07-88	US-A-	4817149	28-03-89
		CA-A-	1301660	26-05-92
		DE-D-	3850417	04-08-94
		DE-T-	3850417	13-10-94
		JP-A-	63224600	19-09-88
WO-A-9113430	05-09-91	AU-B-	649029	12-05-94
		AU-A-	7487591	18-09-91
		CA-A-	2051386	29-08-91
		EP-A-	0470245	12-02-92
		JP-T-	4505670	01-10-92
		US-A-	5148489	15-09-92